L Number	Hits	Search Text	DB	Time stamp
1	39	adsl and atu and cyclic	USPAT	2004/01/02 16:07
2	0	adsl adj8 atu adj8 cyclic	USPAT	2004/01/02 16:02
3	0	adsl adj15 atu adj15 cyclic	USPAT	2004/01/02 16:02
4	0	adsl and atu and (cyclic adj2 sequence)	USPAT	2004/01/02 16:07
5	0	(cyclic adj2 sequence) and modem and (echo adj2 cancellation) and fft	USPAT	2004/01/02 16:09
6	0	(cyclic adj2 sequence) and modem and (echo adj2 cancellation)	USPAT	2004/01/02 16:08
7	1	("5917809").PN.	USPAT	2004/01/02 16:49
9	0	(asymmetric adj2 digital adj2 subscriber adj2 line) adj5 cyclic	USPAT	2004/01/02 16:50
10	0	(asymmetric adj2 digital adj2 subscriber adj2 line) adj15 cyclic	USPAT	2004/01/02 16:50
8	181	(asymmetric adj2 digital adj2 subscriber adj2 line) and cyclic	USPAT	2004/01/02 16:53
11	3	(echo adj2 cancellation) and ((cyclic or cyclical) adj2 sequence)	USPAT	2004/01/02 17:00
12	34	(echo adj2 (cancellation or canceling or cancel or cancelled or cancels)) and (fft or (frequency adj1 domain)) adj2 (coefficient)	USPAT; EPO; DERWENT	2004/01/02 17:32
13	57	(echo adj2 (cancellation or canceling or cancel or cancelled or cancels)) and (fft or (frequency adj1 domain)) adj8 (coefficient)	USPAT; EPO; DERWENT	2004/01/02 17:59
14	22	(echo adj2 (cancellation or canceling or cancel or cancelled or cancels)) and (fft or (frequency adj1 domain)) adj8 (coefficient) and (iteration or iteratively)	USPAT; EPO; DERWENT	2004/01/02 18:00
15	22	(echo adj2 (cancellation or canceling or cancel or cancelled or cancels)) and (fft or (frequency adj1 domain)) adj8 (coefficient) and (iteration or iteratively)	USPAT; EPO; DERWENT	2004/01/02 18:06
16	3	(echo adj2 (cancellation or canceling or cancel or cancelled or cancels)) and (fft or (frequency adj1 domain)) adj8 (coefficient) and (iteration or iteratively) and (cross adj2 correlation)	USPAT; EPO; DERWENT	2004/01/02 18:06
17	3	(echo adj2 (cancellation or canceling or cancel or cancelled or cancels)) and (fft or (frequency adj1 domain)) adj8 (coefficient) and (iteration or iteratively) and (filter adj2 length)	USPAT; EPO; DERWENT	2004/01/02 18:06

Search 12

US-PAT-NO: 6628704

DOCUMENT-IDENTIFIER: US 6628704 B1

TITLE: Equalizer training for ADSL transceivers under TCM-ISDN crosstalk environment

---------- KWIC ---------

Detailed Description Text - DETX (18):

FIG. 2a illustrates an exemplar timing diagram for a TCM-ISDN line in accordance with the present invention. During time period or window 22, data is output from a TCM-ISDN transceiver at the central office to the remote ISDN transceiver at the customer's premises. This data arrives at the remote transceiver during reception window 24. A pause occurs when no data is transmitted. This pause is sometimes called the turnaround period. During period 26, upstream data is transmitted from the remote transceiver to the central office transceiver during window 28. At any particular time, only one end of the TCM-ISDN line is transmitting, while the other end is receiving. Echo cancellation is not needed since the echo of the transmitted signal does not have to be removed. While such a TCM-ISDN system is effective for reducing cross-talk in the TCM-ISDN system itself, it is difficult to add newer ADSL systems in the same cable bundle because of the cross-talk from the ISDN lines.

Claims Text - CLTX (22):

22. The method of claim 1, wherein the step of selectively training at least one of the frequency-domain equalizers during the first training signal

sequence further comprises selectively updating a set of frequency-domain
equalizer coefficients based on error signals generated by that
frequency-domain equalizer and symbol decision.

Claims Text - CLTX (23):

23. The method of claim 1, wherein the step of selectively training at
least one of the frequency-domain equalizers during the second training signal
sequence further comprises selectively updating a set of frequency-domain
equalizer coefficients based on error signals generated by that
frequency-domain equalizer and symbol decision.

Claims Text - CLTX (25):

25. The method of claim 24 wherein the step of selectively updating at
least one of the frequency-domain equalizers further comprises selectively
updating a set of frequency-domain equalizer coefficients based on error
signals generated by that frequency-domain equalizer and symbol decision.

US-PAT-NO: 6535552

DOCUMENT-IDENTIFIER: US 6535552 B1

TITLE: Fast training of equalizers in discrete multi-tone (DMT) systems

---------- KWIC ---------

Abstract Text - ABTX (1):

A method for fast training of equalizers in a DMT system begins by normalizing the incoming receive signal (.nu.) via steps (108-116). By normalizing the signal, the convergence rate of the training algorithm becomes relatively independent of channel line length so that long line lengths may converge to optimal equalizer coefficients in short time periods. The method also iteratively adjusts the filter coefficients w over time by using an adaptive gain vector .mu. that is updated on a component-by-component basis on each iteration via steps (114-146). By allowing each component of the vector .mu. to iteratively adapt independent of all other components in the vector .mu. based upon the binary sign bit of both real and imaginary components of frequency domain gradient vectors G, a convergence to optimal equalizer filter coefficients will occur in a short period of time.

Detailed Description Text - DETX (34):

The FLMS routines and/or hardware may also be used in frequency domain echo canceller (FREC) circuitry and/or like software methods. The frequency domain echo canceller (FREC) is an important application for DMT transceivers that implement spectral overlap. Spectal overlap is when a

downstream and upstream
signal from a tranceiver is communicated over an overlapping frequency band.
Spectal overlap is desired since it reduces the maximal frequency of the
communication system whereby even longer line lengths can be supported. The
reduction in the maximal frequency communicated on the line allows longer line
lengths since the parastitics of the channel are usually a direct function of
frequency (i.e., the higher the frequency, the worse the parasitics). In FREC,
the idea is to explore the frequency and time domain of the DMT data and design
a frequency domain update of the adaptive FREC filter coefficients. An
efficient method for FREC adaptive design was disclosed by Ho et al. U.S. Pat.
No. 5,317,596, issued on May 31, 1994, and entitled "Method and Apparatus for
Echo Cancellation with Discrete Multitone Modulation." Similarly, the actual
update of the FREC filter coefficients may be based on the FLMS algorithm with
a similar structure of the FLMS TEQ training taught herein. Using an adaptive
step size scalar or vector .mu. improves the echo cancellation performance
because the training process will present optimal convergence rates for every
carrier/bin and/or provide improved numerical stability. Therefore, there are
many practical applications for the FLMS algorithms taught herein.

Claims Text - CLTX (34):
34. A method for training a receiver, the method comprising the steps of:
(a) Receiving a time domain receive signal from a communication channel; (b)
finding a maximum absolute value of the time domain receive signal over a
specific period of time; (c) normalizing the time domain receive signal to a
normalized receive signal using the value derived from the maximum absolute

value of the receive signal; (d) transforming the time domain receive signal to a frequency domain receive signal; (e) providing a frequency domain initialization sequence in the receiver; (f) calculating a frequency domain target impulse response from the frequency domain receive signal, the frequency domain initialization sequence, and adaptive filter coefficients; (g) transforming the frequency domain target impulse response to a time domain target impulse response; (h) finding a window of maximum energy in the time domain target impulse response; (i) transforming the window to a frequency domain window; (j) finding an error using the frequency domain receive signal, the frequency domain initialization sequence, the frequency domain window, and adaptive filter coefficients; (k) finding a gradient using the error and the frequency domain receive signal; (l) updating a step size vector using sign changes of real and imaginary entries in the gradient; (m) using the step size vector and the gradient to update the adaptive filter coefficients; (n) transforming the frequency domain of the adaptive filter coefficients to the time domain to create time domain adaptive filter coefficients; (o) finding a window of maximum energy in the time domain adaptive filter coefficients; (p) transforming the window to the frequency domain adaptive filter coefficients; and (q) repeating steps (a) through (p) a number of times to obtain final filter coefficients for use by the receiver.

US-PAT-NO: 6101864

DOCUMENT-IDENTIFIER: US 6101864 A
See image for Certificate of Correction

TITLE: Method and apparatus for generation of test bitstreams and testing of close loop transducers

---------- KWIC ---------

Detailed Description Text - DETX (9):
FIG. 2B illustrates a preferred embodiment of the conceptual method of generating a one bit .SIGMA..DELTA. modulated test signal. A microprocessor 100 is provided with a memory section 60' which, through algorithms or look-up tables, generates a 24 bit pulse code modulated test signal when a particular type test signal is input. Such test signals may be, for example, a sine wave, a dual tone signal, a multi-tone signal and so on. The arrangement of FIG. 2B may also be embodied in a microprocessor/FPGA arrangement, a microprocessor/ASIC arrangement or in an ASIC.

US-PAT-NO: 6618480

DOCUMENT-IDENTIFIER: US 6618480 B1

TITLE: DAC architecture for analog echo cancellation

---------- KWIC ---------

Abstract Text - ABTX (1):

Echo cancellation in data communication between modems utilizes analog echo cancellation to lessen reduction of usable dynamic range of the received signal at the input to the analog-to-digital converter (DAC) in the receiver. Two digital-to analog (D/A) conversions are provided in the modem's analog front end (AFE). One generates the analog signal for transmission. The other generates an analog representation of a cancellation signal that is used to electronically cancel the echo before analog-to-digital (A/D) conversion of the received signal. A preferred embodiment utilizes multiplexed DAC architecture to emulate two DACs by sharing DAC circuitry between data paths of the two D/A conversions.

TITLE - TI (1):

DAC architecture for analog echo cancellation

Brief Summary Text - BSTX (9):

Because the transmitter and receiver are co-located in the same transceiver of a wireline modem, the receiver portion can exploit knowledge of the transmitted signal to extract the reflected portion of it from the receive path. Algorithmic echo cancellation techniques can be applied to first

track-and-hold
clock signals in FIG. 15.

Detailed Description Text - DETX (20):

A preferred technique for resolving problems caused by the T/2 time offset is to compensate for the offset in the signal processing algorithms that are digitally implemented in wireline modems. For modem applications, the second path is used for analog echo cancellation. Therefore, the signal synthesized by the echo canceler will have the half-sample delay. The half-sample delay can be considered during analysis of the echo path through the hybrid and calculation of the model parameters used to cancel the echo. For example, consider the cancellation signal c(t) that is produced when the hybrid model h[n] is convolved with the transmit symbols and passed through a single DAC. Also consider the cancellation signal c.sub.m (t) produced from the multiplexed DAC, and a different hybrid model h'[n]. If the multiplexed DAC were used with h'[n]=h[n], then the resulting cancellation signal would be c.sub.m (t)=c(t-T/2). One could achieve c.sub.m (t)=c(t) by upsampling the DAC input by a factor of two, performing near-ideal interpolation between samples, advancing the signal by one sample, then downsampling to produce the correct DAC input. While this procedure produces the desired result, a more straightforward solution is to incorporate knowledge of the T/2 delay into the model parameter computation.

Detailed Description Text - DETX (21):

An example of how to modify the model parameter computations for a typical wireline modem is given here. A large class of model computation algorithms use a frequency-domain performance criteria and adaptation

algorithms to adjust
the time-domain coefficients until some minimum level of performance is
achieved. A conventional training algorithm that is used to form a model for
the hybrid echo path in wireline modems uses a least-mean-square (LMS)
algorithm to adapt the time-domain coefficients to force the frequency-domain
response of the model to approach a desired response. When the time-domain
coefficients form a filter that will be convolved with the signal that goes out
through the multiplexed-DAC path with the T/2 delay, the delay can be
pre-compensated by including an inverse effect in the frequency-domain design
criteria. If the model estimation algorithm for a single DAC attempts to
converge to a frequency-domain representation of H[k], then the target
frequency response can be modified to be
H[k]e.sup.-j2.sup..sup..pi. .sup.f
T/2. The exponential term adds a linear phase component of -2.pi. f T/2 to
the prior target frequency response. Multiplication by this linear phase term
results in the inverse frequency response having an advance of one-half sample.
It will appear as though the upsample, interpolate, advance, and resample had
been applied.